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ACOUSTIC SCALING: instrumentation

No. 1972/34

Research Department, Engineering Division
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RESEARCH DEPARTMENT

ACOUSTIC SCALING: INSTRUMENTATION

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ACOUSTIC SCALING: INSTRUMENTATION

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ACOUSTIC SCALING: INSTRUMENTATION

Summary

Previous reports in this series have described the principles of acoustic modelling and the success of the proving experiment. This success has depended substantially on the instrumentation employed in the model. In this report details are given, for example, of the methods used to enable an adequate signal-to-noise ratio of the final recording to be obtained, a high degree of reproduction to be achieved by the loudspeakers and a rapid drying cycle to be maintained.

1. Introduction

Details have been given in a previous report[1] of the principles controlling acoustic modelling. Briefly, a model of one-eighth linear dimension has been made of an existing studio and the frequencies used in it are multiplied by 8, i.e. they range from 400 Hz to 100 Hz, to keep the ratio of wavelength to dimensions correct. In addition to carrying out objective tests, such as reverberation tlme, stereo music free of reverberation is played at eight times normal speed into the model and recorded by miniature microphones on a special tape recorder. The recording is then played back at normal speed and the resulting audio-frequency programme can be compared with that recorded in the real studio.

In order to carry out these tests it was necessary to design, or modify, a number of items of equipment such as loudspeakers, microphone head-amplifiers and magnetic recording tape machines and the subjective results of the final experiment were critically dependent on the equipment involved. For example, the overall signal-to-noise ratio, which must be good enough to allow of meaningful subjective assessments of the acoustic qualities of the model, depends on the background noise levels of the microphone preamplifiers, of the magnetic tape recorder, and on the sound levels which can be generated by the loudspeakers; the sound quality itself depends profoundly on the performance of the loudspeakers especially from the point of view of colouration. This report gives details of the requirements, design and performance of such items with the exception of the model reverberation room which forms the subject of a seperate report[2].

2. Microphone preamplifier

This section describes the microphone preamplifier, which was designed and built within the Section as no commercial version had an adequate signal-to-noise ratio.

To pick up the sounds covering a bandwidth extending from 400 Hz to 100 kHz, no practical alternative exists to a condenser microphone assembly, consisting of a microphone capsule and its associated preamplifier.

The actual value of the capacitance of such a capsule is of the order of 6 pF for the 6 mm unit used in the model; at the time work was commenced this was the smallest known unit. To preserve strictly the scaling factor of 1 : 8 used elsewhere in the model, a microphone with a diaphragm diameter of 3 mm would be required although its lack of sensitivity and therefore poor signal-to-noise ratio would weigh heavily against its use.

As the signal source of a condenser capsule appears in series with such a very small capacitance, it is necessary to feed its output into a very high resistance in order to avoid both attenuation of low-frequency signals and generation of electrical noise. To avoid direct attenuation of the signal it is essential to avoid the capacitative shunt effect of a cable connection between capsule and preamplifier. However, it is equally essential to screen efficiently the connection between capsule and preamplifier to avoid hum and spurious external pickup in the high impedance involved. For these reasons it is universal practice to embody a condenser microphone capsule as part of the preamplifier housing.

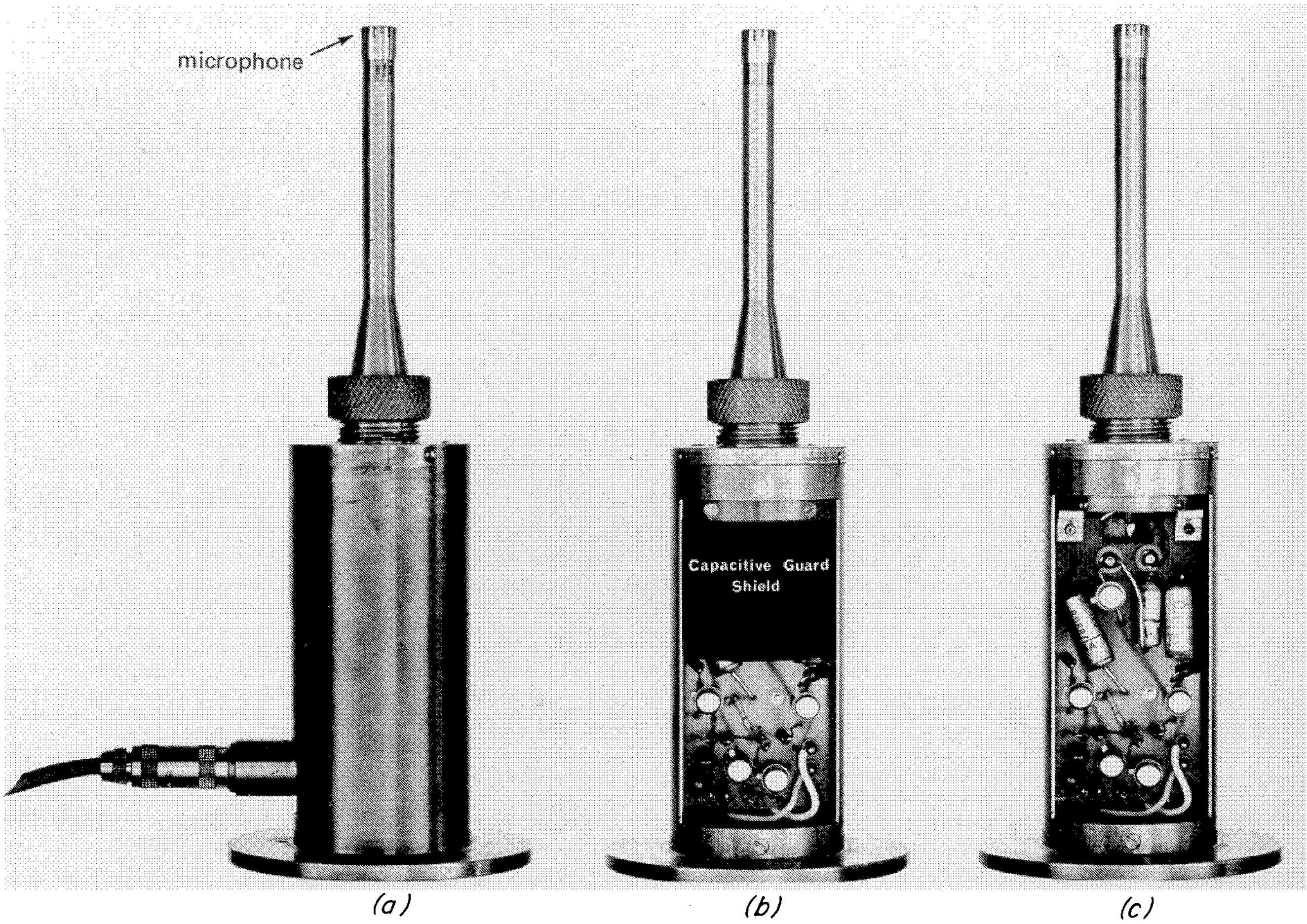


Fig. 1 - Appearance of 6 mm microphone and pre-amplifier
(*a*) External (*b*) Internal, with guard shield (*c*) Internal, without guard shield

The essential requirements of the preamplifier were as follows:

It should introduce the minimum possible electrical noise with the system.

The input impedance should be as high as possible and certainly not less than 100 megohms resistance with a shunt capacitance of not greater than 1 pF. A uniform frequency response was required extending from 400 Hz to 100 kHz.

The gain should be in the order of 12 dB with an output impedance suitably low to feed a following amplifier separated by several yards of cable.

Fig. 1 *(a)* shows the mechanical design of the preamplifier. The overall height of the unit was arranged to be 6 inches (15 cm), thus the capsule would be at an effective height of 48 inches (120 cm) above the floor in a full scale hall. This is approximately 'ear height' for an average seated adult.

The capsule is screwed into a thin rigid tapered extension tube to isolate it acoustically as far as possible from the main body of the preamplifier. It was of paramount importance that the best possible signal-to-noise ratio was obtained in the preamplifier and as both input capacitance and resistance bear directly on this, considerable attention was directed to them.

There already exists an extensive literature on the design of valve preamplifiers for condenser microphones, (3, 4, 5,) and more recently the use of the very high input impedance of the Field Effect transistor has become fairly well-established (6, 7, 8,) in the normal audio frequency range. For the preamplifiers used in the model, the F.E.T. offered clear advantages in size and performance.

Fig. 2 shows the circuit used. The input capacitance of the amplifier is made up from the sum of the gate/drain and gate/source capacitance and the wiring and components capacitances were effectively neutralised by enclosing all the components and wiring in the dotted area of Fig. 2 within a shield which extended coaxially from the base of the microphone capsule down into the body of the preamplifier. Fig. 1 *(b)* and *(c)* show the unit with and without shields.

For such a shield system to achieve its maximum effect, it must be driven from a signal source of the same amplitude as the original input signal. This infers the use of an efficient source follower.

The F.E.T. at the input, TR1 a BFW56, is connected as shown with TR2, a 2N2484, used as a source load. This forms a load with a high a.c. impedance and also ensures that the current through the F.E.T. is held at a sufficiently high level to keep the gain well maintained

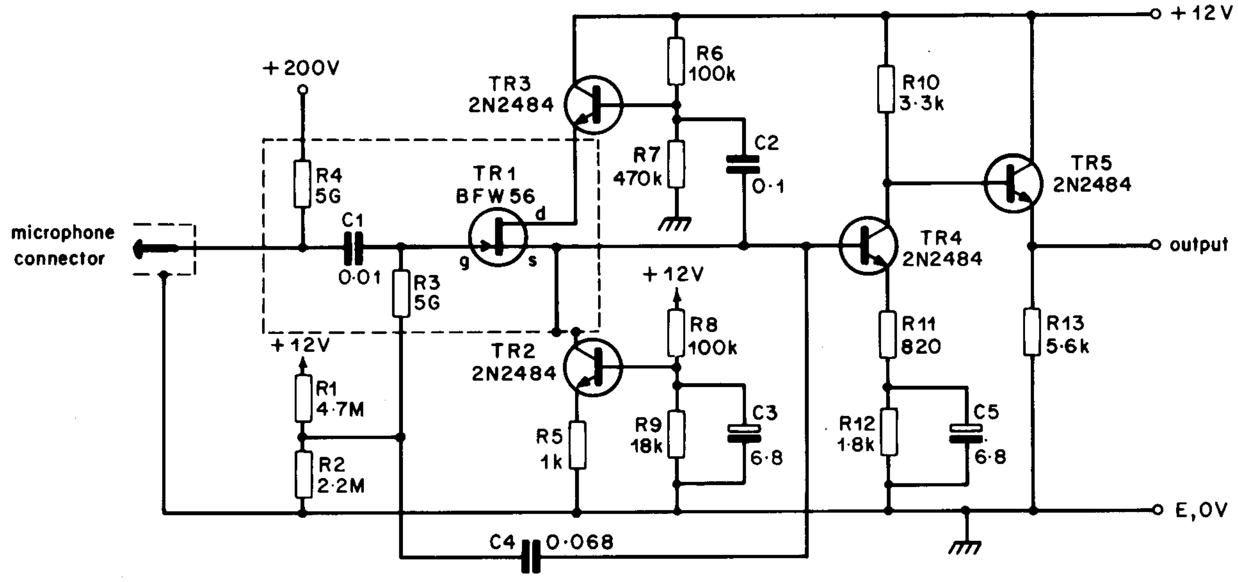


Fig. 2 - Circuit diagram of microphone pre-amplifier

and therefore the noise figure low. With this arrangement, an output to input ratio of 0·99 : 1·0 is obtained. The capacitive shields are fed from the junction of TR1 and TR2.

Connecting the F.E.T. as a source follower virtually eliminates the gate to source capacitance, leaving only the gate to drain capacitance. This also is neutralised by applying the same a.c. signal to the drain of the F.E.T. via TR3. By these means the total input capacitance was reduced to less than 0·3 pF.

The noise introduced by the preamplifier is produced by two sources:—

(i) that generated by the F.E.T. itself which is governed by the particular type of F.E.T. chosen and will vary somewhat with current and also from specimen to specimen of the same type, but will generally have a spectrum of equal energy per cycle over the frequency of interest.

(ii) that which is intrinsic in the value of the input circuit components themselves which in effect has a '$\frac{1}{f}$ spectrum'.

The noise generated by the F.E.T. is obviously optimised by the choice of transistor type and for this purpose the Texas Instrument BFW56 appeared to have the best available compromise between input capacitance, mutal conductance and noise.

It may be shown that the fundamental noise generated by the input circuit of Fig. 2 is, for a given value of capacitance (in this case the 6 pF of the capsule plus the 0·3 pF of the amplifier) solely dependent on the resistive component in shunt with it. The higher the value of resistance the lower the noise.

A practical limit has clearly to be set for the upper value of the resistance that may be put in series with the gate electrode. In the case of the BFW56, the gate leakage current at room temperature was sufficiently low to permit a value of 10,000 megohms without impairment of stability. Even higher values would probably be possible but for difficulties associated with surface leakage.

To reduce surface leakage to a minimum, as few as possible mechanical securing points were used in the input stage and where such points were unavoidable, PTFE supports were used. It was observed, incidentally, that problems with leakage occurred whenever soldering had been carried out in the vicinity of these supports. It was found that a fine film of resin flux (probably bearing solder particles) had vaporised on to the PTFE so degrading the insulation.

TR4 forms a simple amplifier and TR5 provides an emitter follower output stage.

The final performance achieved was as follows:

Input impedance:	Greater than 3000 megohms in parallel with less than 0·3 pF.
Unweighted Noise:	−103 dB relative to 1 volt, with input terminated in 6 pF, and a bandwidth 400 Hz to 100 kHz.
Equivalent Unweighted Sound Noise Level:	41 dB w.r.t. 20μ N/m^2
Gain:	12 dB
Frequency response:	± 0·5 dB, 400 Hz to 100 kHz
Output impedance:	40Ω

The noise spectrum was measured and as hoped for consisted of equal energy/cycle, showing that the compon-

ent from the input impedance was adequately low, and also that the $\frac{1}{f}$ noise from the F.E.T. was negligible.

The noise figure of − 103 dB is about 15 dB lower than that of the commercial preamplifiers available.

3. Loudspeakers

a) General

The performance of the loudspeakers affects the subjective assessment of the model acoustics in a direct way and several different qualities are demanded of them. They are therefore considered in some detail.

Firstly, a sufficiently high sound level must be produced in the model so that, in conjunction with the microphones used, the signal-to-noise ratio produced is adequate to prevent electrical background noise obscuring subjectively the acoustics of the model. Secondly, not only must the response/frequency characteristic cover the whole spectrum uniformly, but in addition it is of the highest importance to avoid any substantial colouration of the sound produced in the model, otherwise the assessment of the acoustic properties of the model will be affected. Finally, the ratio of direct to reverberant sound is directly dependent on the directional properties of the loudspeakers. These points will now be discussed separately.

b) Directional properties

Taking the last requirement first, as this affects the whole design, various directivities for model loudspeakers have been suggested by other workers in the field; examples are that of the human voice[9], an omidirectional source[9], or at the opposite extreme an extended source intended to simulate an orchestra. Now in practice most of the musical instruments in an orchestra each cover only a restricted frequency range and acoustically are in random phase; an extended source therefore would not imitate the directional properties of an orchestra at all well, whilst on the other hand separate recordings for each instrument would involve many separate amplifiers and loudspeakers, an effort which would not be worth while, in the first place at any rate.

It would be difficult to make a truly omidirectional source which covered such a wide bandwidth and the most practical solution for a proving experiment seems to be to make two monitoring type loudspeakers of the multi-unit type to cover the whole frequency range. This method has the advantage that an existing stereo recording[10] of non-reverberant music can be replayed through a pair of loudspeakers, having a similar type of construction, in the real studio and thus a close subjective comparison of the acoustics of the model and studio can be made.

As already mentioned the frequencies to be covered in this case are eight times the audio frequency band and therefore range from 400 Hz to 100 kHz and in the first instance the frequency band was split into two. Any bass units large enough to radiate substantial amounts of power at the lowest frequency of interest, 400 Hz, are rather directional at the upper end of their frequency band. Therefore, to ensure an adequate off-axis response at the top of the band handled by the low frequency units, two units were employed with their axes inclined at an angle of about 60° to each other in the horizontal plane, thus radiating over a total included angle of about 120°. For the high frequencies the only units that are commercially available are of the electrostatic type made by Müller BBN in Germany. These have a slack diaphragm of sputtered polyethylene terephthalate about 6 μm thick; the back electrode consists of a series of concentric etched annuli thus forming a design intermediate between the Sell and the open electrode types. The diameter of each unit is about 2·5 cm and as the wavelength of sound is only 0·33 cm at 100 kHz, the units are about eight wavelengths in diameter, and the radiation from each unit is therefore extremely directional at this frequency. In order to obtain both an overall sound level and a directivity comparable with those of the low frequency units a large number of h.f. units were so mounted on a hemisphere that they also covered an included angle of about 120°.

c) Tonal quality

Taking the second requirement next, the frequency range covered by the h.f. units extends from just below 10 kHz to 100 kHz. Unfortunately our tests had shown that the h.f. units will not accept full modulation below about 15 kHz, and this, therefore, had to be accepted as the cross-over frequency. As no commercial unit covered the low frequency range required a special unit 11 cm in diameter and with a cone of 7 cm diameter was designed using 0·025 cm thick Bextrene for the cone and having a pvc surround. To cover the required bandwidth the cone was made highly flared and the consequent tendency to an air mode resonance inside the cone was eliminated by means of a flat central diaphragm of p.v.c. This diaphragm also has the effect of improving the off-axis response/frequency characteristics. The units were first mounted in a closed cabinet, volume 0·003m^3 of 18 SWG steel damped with a 6 mm layer of polyvinyl acetate damping compound.

As mentioned above, the h.f. units were mounted on a hemisphere. This is 10 cm in radius and is made of thin copper so that resonances are well damped. As a precaution against air modes of resonance the space between the units and the hemisphere was filled with foamed polyurethane and the hollow rear of the hemisphere was also filled with the same material.

The h.f. units are capacitive in impedance, about 85 pF each, and need a 400 V polarizing potential so it was not convenient to design a high pass network as a crossover filter to operate directly into the units in the conventional way for moving coil units; the network was therefore placed ahead of the power amplifier which supplied them with bias and with the modulation signal.

The quality of sound reproduction was tested by radiating programme from the complete loudspeaker in the free field room and recording it at high speed on the tape

recorder. This was then played back at one eighth the speed through a type LS 5/5 monitoring loudspeaker in a listening room and the direct and recorded quality were compared.

It was found that the sound reproduction was coloured to a considerable extent mainly in the lower and middle frequency range. This corresponds to frequencies up to 1·5 kHz, or about 12 kHz as reproduced by the model loudspeakers and thus involves the 'low frequency' units.

Now it becomes obvious that the demands made on the model loudspeakers are far more stringent than those required of loudspeakers in the normal audio frequency range. If, for example, a monitoring loudspeaker had a resonance at 8 kHz with a decay time of 10 ms, this would probably be inaudible as a colouration. Such a resonance in the model loudspeaker, however, would, when the tape recording was slowed down by a factor of 8 to 1, be reproduced at 1 kHz, i.e. in the most sensitive audio frequency region, and with a decay eighth times as long, i.e. 80 ms, and this would most certainly be highly disturbing. The transient properties of the model loudspeakers must therefore be very much better than the best of those used for normal monitoring. The performance of loudspeakers, which has taken fifty years to reach its present standard of quality, has therefore to be radically improved for use in the model.

The two frequency band loudspeaker system was therefore scrapped and replaced by a three frequency band loudspeaker system shown in Fig. 3. The single 11 cm unit was used for the bass up to a frequency of 3 kHz (i.e. 375 Hz referred to normal frequencies) as the angle of radiation was adequate up to this frequency. Above this a K.E.F. high frequency unit type T27 was used for the 'middle frequencies' up to 21 kHz. This is a moving coil unit and has a thin domed diaphragm about 2·5 cm diameter made of polyethylene terephthalate; tests showed it to be remarkably free of colouration. Above 21 kHz the electrostatic units took over as before, extending the range to 100 kHz.

With this arrangement, as with three way conventional speakers, no unit is stretched beyond the engineering limits and much more uniform response characteristics result. The sound quality is also a definite improvement on that of the two way system, and in particular the colouration at 800 Hz is absent.

The equalisiation of the loudspeaker is a difficult problem as the sound quality is by no means entirely determined by the steady state response/frequency curves. The method adopted was to replay at high speed through the loudspeaker in the free field room a tape containing a variety of types of programme, the sound output being picked up by a microphone situated on the loudspeaker axis. On replay at normal speed the sound quality was compared with that from the original tape starting with the objectively determined equalisation, which was then adjusted by ear until the sound quality was as close as possible to the original. The difference between the two equalisations, though significant, was only small,

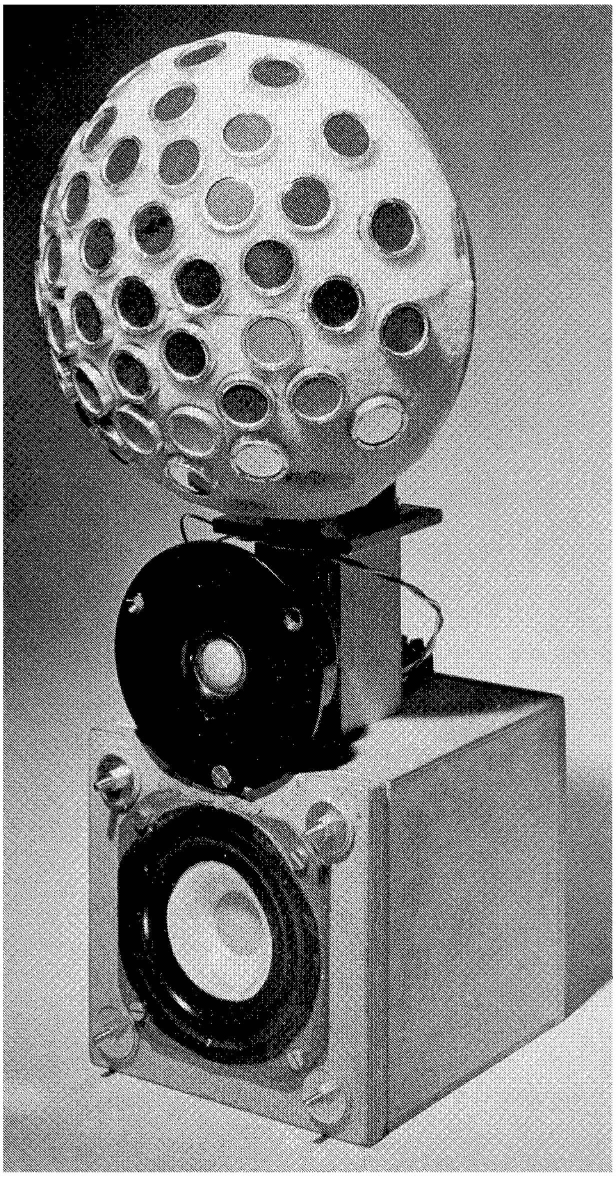


Fig. 3 - Appearance of three frequency-band loudspeaker system

thus indicating the comparative freedom from colouration of the model loudspeaker, even under these stringent conditions.

A curve of the axial response/frequency characteristic for the complete loudspeaker with crossover network, but unequalised, is shown in Fig. 4. Even in this condition it is seen to be within ± 5dB of a uniform output over most of the frequency range.

d) Power handling capacity

Finally, taking the third requirement for the loudspeakers, the sound level required in the model depends on the equivalent noise level of the microphone and the signal-to-noise ratio of the tape recorder because the noise generated by the loudspeakers is negligible. The former was shown in the previous section to be 41 dB with respect to 20 $\mu N/m^2$, the latter was measured to be 52 dB.

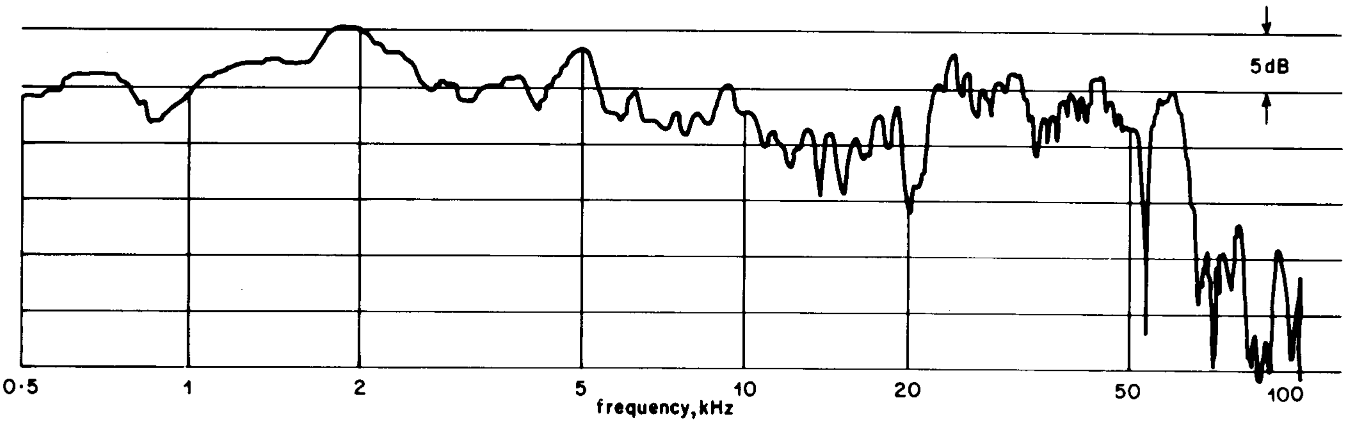


Fig. 4 - Unequalised axial response/frequency characteristic of loudspeaker system

For a final signal-to-noise ratio not appreciably worse than that of the recorder the sound level to be generated in the model should be 99 dB with respect to 20 $\mu N/m^2$.

Now the sound level (L) in a reverberant enclosure is given[11] by

$$L = H + 10 \log_{10} T - 10 \log_{10} V + 14 \text{ dB}$$

where the energy input $H = P + 10 \log_{10} 2\pi R^2$

T = reverberation time (secs)
V = volume of enclosure (m^3)
P = sound pressure (dB w.r.t. 20 $\mu N/m^2$)
R = radius (m)

In our case the full scale studio is about 6,250 m^3 in volume and reverberation time is 1·8 secs.

The low-frequency units each generate a sound pressure of 73 dB with respect to 20 $\mu N/m^2$/volt at a radius of 1·5 and the impedance is 12 Ω. The power rating is 10 watts for sine wave or 20 watts for programme material and with the two loudspeakers the maximum sound level in the one eighth scale model is therefore 98·5 dB continuous or 101·5 dB for programme. For frequencies within the range of this unit a signal-to-noise ratio for programme of 60 dB is therefore possible in the latter case.

The middle frequency units have a mean sensitivity of 70 dBwrt 20 $\mu N/m^2$/volt at a radius of 1·5 m and have an impedance of 6 Ω. The power rating is again 10 watts for sine waves or 20 watts for programme and with the two units used in the model a signal-to-noise ratio for programme of 55 dB is therefore possible over the frequency range covered by this unit.

The individual h.f. units are stated by the manufacturer to deliver about 10^{-4} watts at 15 kHz rising by about 7 dB at 50 kHz and falling to 7 dB below 10^{-4} watts at 100 kHz. The low-frequency efficiency at 20 kHz is increased by about 20 dB by the proximity of the other units when mounted on the hemisphere. If therefore we take the value at 50 kHz we have a potential signal-to-noise ratio, for a total of 90 units in the two loudspeakers, of 75 dB falling off at 100 kHz to 47·5 dB. This last figure for signal-to-noise ratio can be appreciably improved by reproducing in the model, programme which is pre-emphasized in accordance with the usual equalisation applied to f.m. broadcasting, and this will yield most benefit just where the signal-to-noise ratio is worst. Thus at 100 kHz, corresponding to 12·5 kHz in real life, the pre-emphasis, and therefore improvement in signal-to-noise ratio, would be about 12 dB, giving a figure of about 60 dB at this frequency.

In view of the signal-to-noise figures just given it may appear that an unnecessary number of expensive* h.f. units are employed. This number was, however, ordered when the first prototypes were built and it was stated by the manufacturer that the sound power radiated from each unit would be some 10 dB lower than the figure given above. Subsequent development, after the order was placed, allowed both the polarizing voltage and the modulating voltage to be substantially increased, thus giving rise to the increase in output of 10 dB mentioned. It will be appreciated that in any case owing to the high directivity of the individual units a large number of them is necessary to give an adequately wide polar diagram.

4. The tape recorder

From the block schematic diagram Fig. 5 it will be seen that the programme fed into the model studio originated in a four channel tape recorder.

The procedure for obtaining a recording for assessment from the model studio was to record the non-reverberant programme in stereo on channels 1 and 2 at 3¾"/sec. (94 mm/sec.). This was then replayed into the model at 30"/sec (0·75 m/sec) and simultaneously the stereo output from the microphones was recorded on channels 3 and 4.

*They cost about £10 each, making a total cost of £900 for the two h.f. groups.

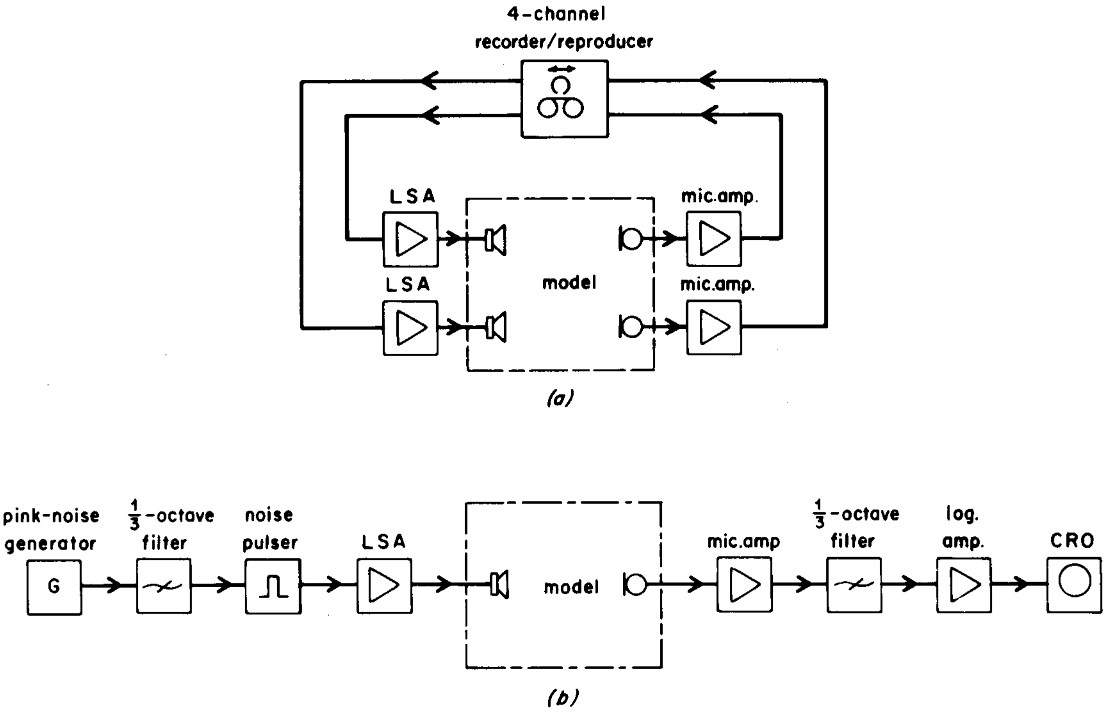


Fig. 5 - Block schematic circuit diagram

(*a*) For music recording
(*b*) For measurement of reverberation time

The tape was then replayed at 3¾"/sec (94 mm/sec) to regain the frequencies at which the programme was originally recorded. The programme now with the added reverberation from the model studio appeared at the output of the replay amplifiers 3 and 4.

Briefly, the requirements of the recorder were as follows:-

> Four channels arranged as two stereo pairs, each with separate record and replay facilities and capable of replaying one pair of channels and recording with the other.
>
> The ability to change speed from 3¾"/sec (94mm/sec) to 30"/sec (0·75 m/sec) and retain a reasonably uniform frequency response from 50 Hz to 12 kHz at 3¾"/sec (94mm/sec) and from 400 Hz to 100 kHz at 30"/sec (0.75m/sec).
>
> A signal-to-noise ratio of not less than 45 dB for 1 pass at 9·5cm/sec or 75cm/sec.

This brief specification, although apparently not very demanding by usual programme standards could not be met by any production machine but could possibly be achieved by modifications made to an instrumentation recorder.

The signal-to-noise ratio for a conventional production machine with 0·05" (1·27 mm) width tracks (i.e. 16 tracks on 1 inch (25mm) tape was 35 dB at a tape speed of 3¾" m/s (9·5cm/s).

In order to improve on this, one inch (25mm) wide tape was still used but the number of tracks was reduced to 4. i.e. the track width was increased to 0·2" (5 mm) which should improve the signal/noise ratio by 6 dB.

As received from the factory the signal-to-noise of the modified machine was disappointing. The original production line test figures of 35 dB was still only just met (by 1 dB) and extensive work was necessary to improve on this figure.

A spectrum of the noise measured at a tape speed of 9·5cm/s for the machine as received is shown in Fig. 6a and the excessive level of the mains hum and its harmonics is evident by the rise in level in the bass.

The hum was identified as that induced in the replay head from drive motors and transformers mounted in close proximity to the head block.

Suitable head screening reduced the hum level by approximately 12 dB.

The head block as supplied has no provision for adjusting the azimuth of the replay head; when this was fitted, the frequency response was improved by 4 dB at 10 kHz, resulting in turn in an improved signal-to-noise ratio.

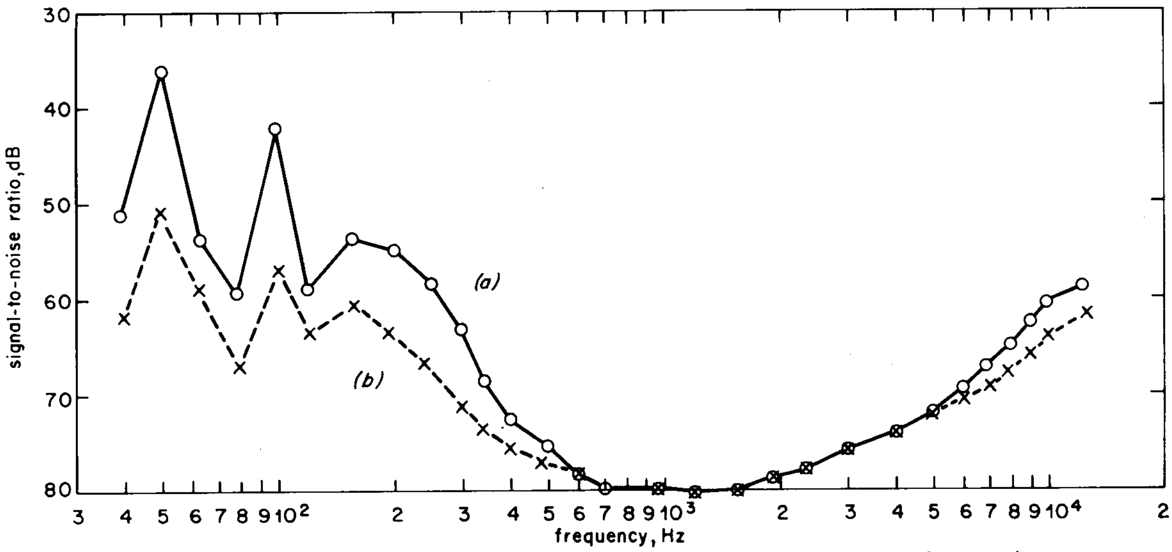


Fig. 6 - Noise spectrum from four-track tape recorder for tape speed of 9·5 cm/s

(*a*) As received (*b*) After modification

The spectrum of the noise after these modifications is shown in Fig. 6b.

The selection of a suitable magnetic tape was complicated by the fact that the standard stereo tape would, in the course of the experiments be played as many as two hundred times. Resistance to wear was therefore a prime consideration, and of the tapes tested, Zonal MB25 was found to combine an unusual degree of resistance to wear combined with low noise and good frequency response.

To obtain an overall frequency response free from cumulative errors it was necessary to equalise the record/replay characteristics at the different speeds with more than usual precision, a protracted procedure when a number of sample tapes were to be tested to decide which one gave the best signal/noise ratio.

The frequency response finally obtained was such that a glide tone when recorded at 3¾"/sec and passed through the low-to-high and high-to-low speed conversion was within ± 2 dB from 50 Hz to 12 kHz. The weighted signal/noise ratio finally achieved was 52 dB.

5. The logarithmic conveter

For some years it has been the usual practice within the section to measure reverberation time both in

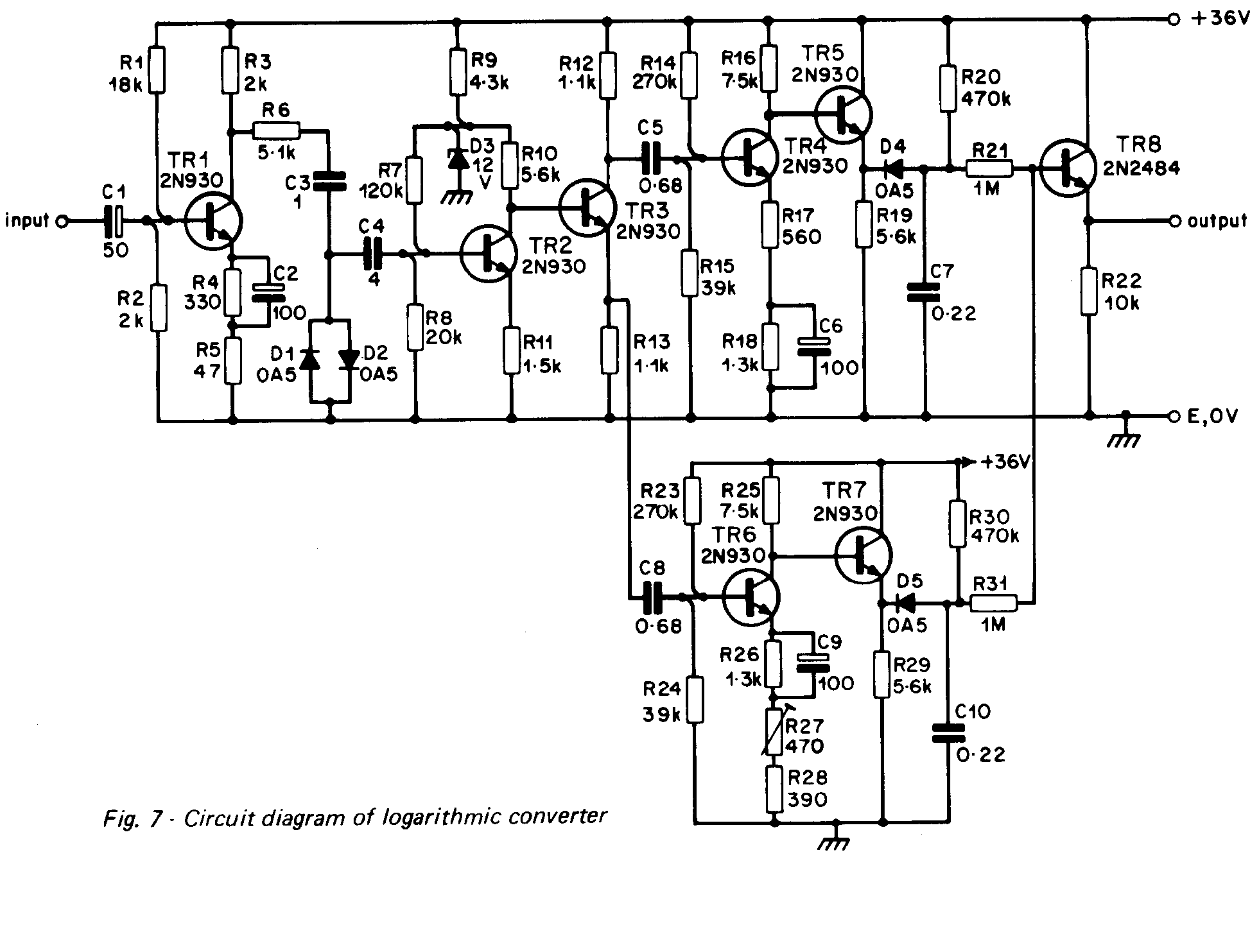


Fig. 7 - Circuit diagram of logarithmic converter

studios and in reverberation rooms by converting the nominally exponential decay of a sound pulse into a linear form by passing the signal through a converter that has an output which is logarithmically related to its input. The slope of the linear decay is directly measurable by applying the output to the Y axis of an oscilloscope and aligning the trace with the cursor of a goniometer.

A new instrument was required for the model work to extend the frequency range to 100 kHz. The basic principle of the new unit is essentially the same as that of the instrument described in Report B056/2, in which use was made of the logarithmic voltage/current relationship inherent in the forward conducting mode of certain semiconductors when fed from a high impedance source.

The new converter uses semi-conductors throughout and the logarithmic elements chosen were germanium gold-bonded diodes in place of the silicon diodes used in the original unit. This choice permitted a substantial reduction in the impedance from which the diodes should be driven, an essential feature to maintain the high frequency response, but reduced the range over which the logarithmic relationship applied to 48 dB. This, however, was judged to be sufficient. The circuit is shown in Fig. 7. TR1 forms a simple high impedance driving stage for the logarithmic elements D1 and D2. The 'logarised' signal developed across them is amplified in Tr2 and fed to a phase splitter Tr3. The outputs of Tr3 are separately further amplified and finally full-wave rectified by D4 and D5. The d.c. output is fed to an oscilloscope input from Tr8 which forms an isolating emitter-follower. The performance is as follows:

> Output voltage is plotted against input level in Fig. 8.
>
> Linearity of slope: better than ± 0·5 dB, 400 Hz to 100 kHz
>
> Minimum rate of decay of signal: 1000 dB/sec

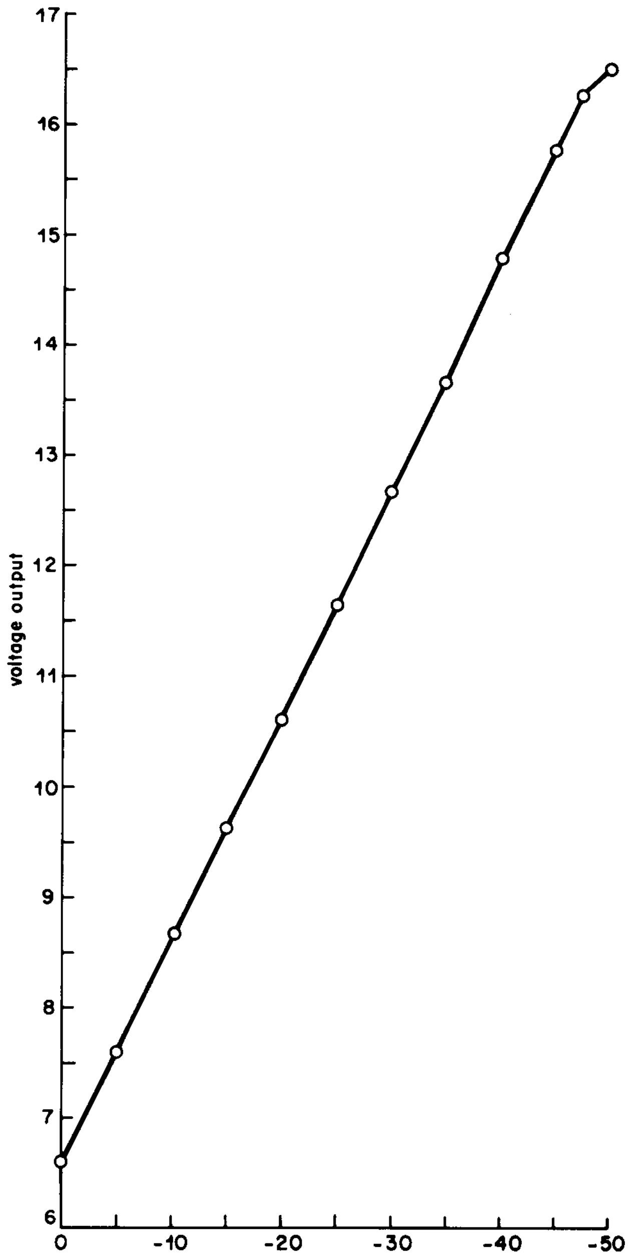


Fig. 8 - Transfer characteristic of logarithmic converter

6. The pink-noise generator

The excitation signals used for the reverberation time measurements in both the model studio and model reverberation chamber[2] were bursts of noise with a bandwidth of one-third octave, the centre points of which were spaced one-third of one octave apart. The noise bursts were formed by feeding the output of a pink noise generator to a bank of 23 one-third octave filters, whose centre frequencies ranged from 400 Hz to 63 kHz, and thence to a gating circuit.

Pink noise, giving equal energy per octave, was used in preference to white noise which produces equal energy per cycle and thus gives a 3 dB per octave increase in level.

Fig. 9 shows the circuit used.

The noise source used in this unit was a Zener diode, the AEI Mr 12.0. No claims are advanced by the manufacturers of these diodes as to their suitability for use as noise sources, but out of about twenty specimens of this diode more than 50% were found to have useful noise outputs. Other manufacturers' products gave very similar results and the type used was chosen on an availability basis only. A Zener current of the order of 40 μA was found to provide a stable noise output although several other comparable outputs were obtained over small ranges of current when it was varied from 30 μA to 200 μA.

The noise output from a diode of this type is essentially white and as the bandwidth of the noise generated may extend to several megahertz it was necessary to provide a simple low pass filter consisting of L_1 and C_2 as a means of band-limiting to avoid overloading succeding stages.

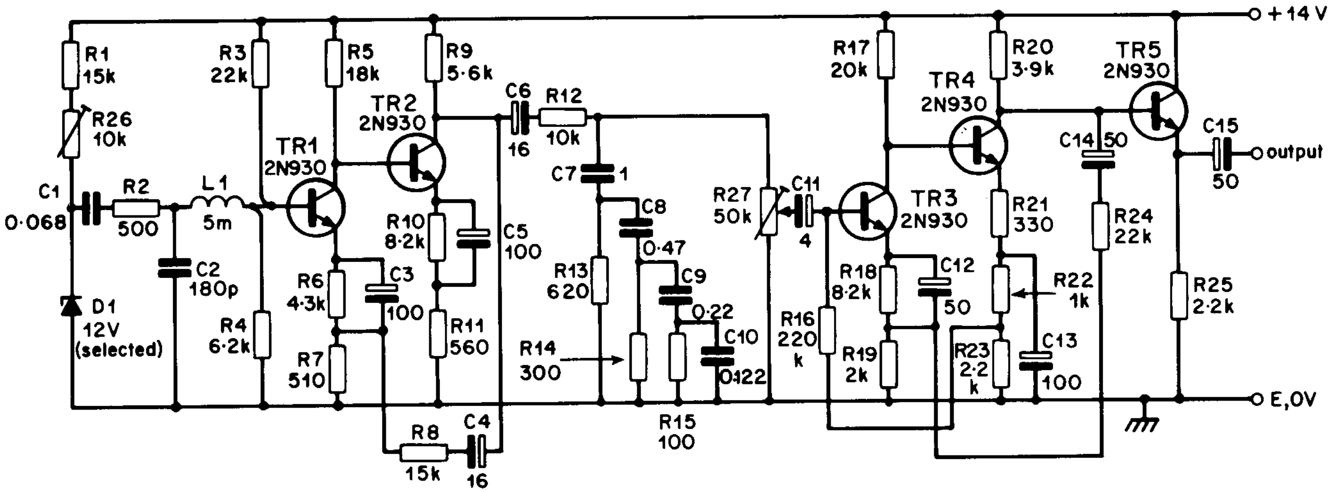


Fig. 9 - Circuit diagram of pink noise generator

The amplified noise is passed through a white to pink filter network having a response falling with frequency at 3 dB per octave. The equaliser consists of C_7 C_8 C_9 C_{10} and R_{12} R_{13} R_{14} R_{15}.

The resultant pink noise is amplified further in Tr_3 and Tr_4 with Tr_5 providing a low impedance output.

Output measured via one-third octave filters:	± 2dB 400 Hz to 63 kHz
Output level:	–12 dB rel 0·775 v r.m.s.
Power consumption:	10 mA at 14 V

7. Drying equipment

In Germany, Spändöck used silica gel for drying his model and therefore this was tried first with the model reverberation room.

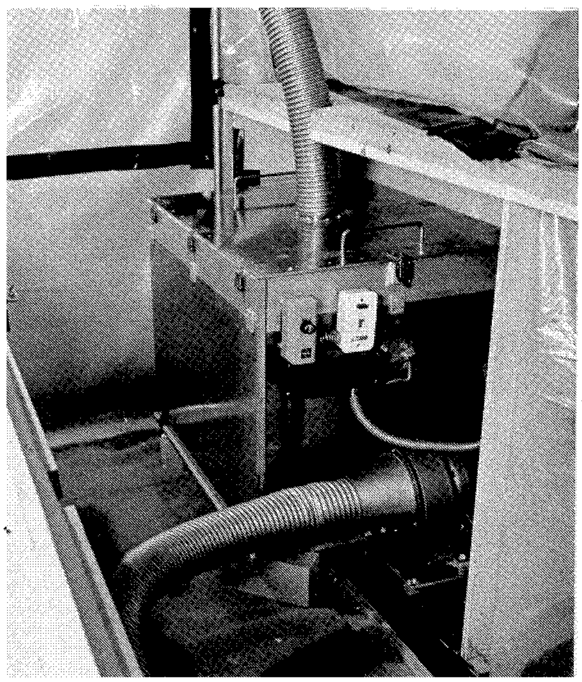
The humidity was first measured with a hair hygrometer having a scale reading down to 1% but suspicions were soon felt as to its accuracy. Subsequent enquiries led to the conclusion that in spite of its scale the instrument was not reliable below 10% relative humidity and that even at this figure, appreciable errors could occur. The instrument was therefore changed for one of the electrical resistance type and this gave more consistent readings at low humidities. Using this more reliable instrument it was found to be very difficult to achieve the required humidity of about 2% and the whole apparatus was very sensitive to small leaks.

The silica gel was therefore replaced by an artifical zeolite and this was found to give much better results, humidities of the order of ½% being easily obtained in the empty reverberation room. Soon after this was obtained it was discovered that the electrical resistance elements were also unstable with time and giving erroneous readings. The hygrometer was therefore replaced by one using capacitive type elements; this has frequently been checked against wet and dry bulb thermometer readings and has shown a high degree of consistency.

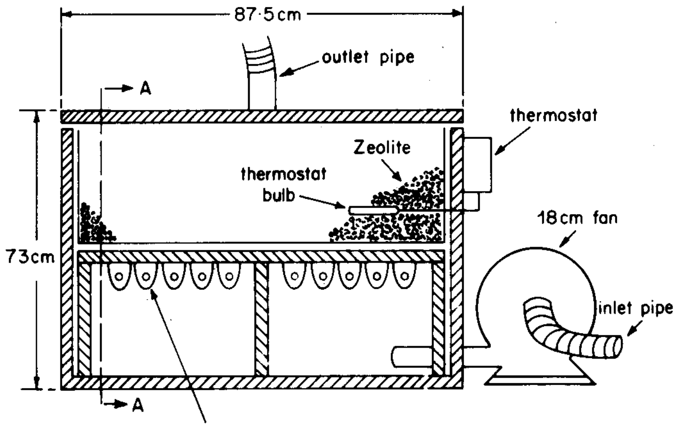
The drying device for the model studio was based on that for the reverberation room but on a correspondingly larger scale (Fig. 10). The tank, size 90 x 70 x 75 cm is divided vertically into two sections by a sheet of perforated brass. The lower portion houses the heating system used for drying the zeolite when not in use. It consists of ten pairs of 1 kW rod type heating elements, as used in electric fires, the two elements of each pair being connected in series to give 500 W at a much lower surface temperature than that normally obtained with a single element. The total heating load is thus 5 kW. The layer of zeolite is 20 cm deep and is held in a receptacle made of wire mesh; imbedded in it near the bottom of the layer is a thermostat bulb for controlling the temperature to 232°C ± 5°C when drying is being carried out. The inlet and outlet air ducts which are of 10 cm diameter are removed during this period. An 18 cm fan delivers air at a 6cm head of water and a volume of 5·5 m^3/min. The circulation of air is sufficient to replace the air in the model studio once every 5 minutes and the humidity of the air leaving the tank is of the order of ½%. Under these conditions the air in the model initially took appreciably less than ½ day to dry out to 4% humidity. Subsequently after the model had been opened for modifications it took only ½ hour to reduce the humidity to this figure and half this time (i.e. ¼ hour) to dry the air between recording sessions. These figures should be compared with that of 'several weeks'[12] for silica gel as used in Germany for the initial drying.

To prevent excessively moist air entering the model when alterations were being carried out, it was enclosed in a polythene tent. The air in the tent was kept at a humidity of about 35% by means of a mechanical dehumidifier and access was by means of a door having an air lock.

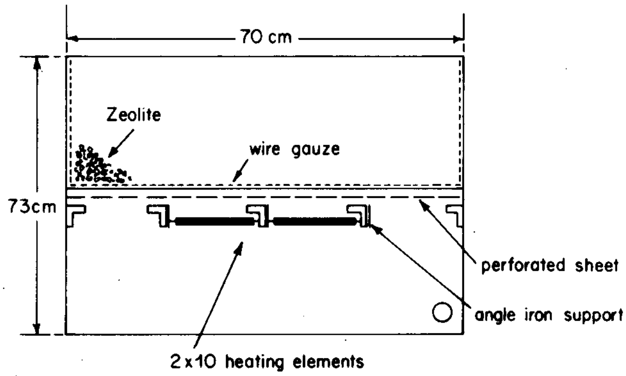
This scheme also had the added advantage that the model, which was made of wood, was not subject to a heavily unbalanced humidity condition between inside and outside which might have caused excessive warpage of the surfaces. No trouble at all was experienced from this cause.



(a)



(b)



(c)

Fig. 10 - Model studio drying apparatus
(*a*) External appearance (*b*) Internal section
(*c*) Section across A-A

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8. Conclusions

Details have been given of the various items of equipment necessary to carry out acoustic modelling. In general it has proved possible to introduce straightforward modifications of existing equipment to satisfy the additional requirements for modelling. The major departure from this rule is that of the loudspeakers and as is so often the case with these transducers, to some extent a rather empirical approach has proved to be necessary. However the final results appear to be satisfactory.

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